

Serial No. 10/674,450

CLAIM LISTING:

This listing of claims will replace all prior versions, and listings, of claims in the application:

1. (Previously presented) A method for processing a voice signal in a communications network, the method comprising:
partially decoding a bit stream corresponding to an encoded version of the voice signal to obtain an excitation parameter corresponding to the voice signal; and
estimating a noise level of the voice signal using the excitation parameter that is obtained directly from the partially decoded bit stream.
2. (Original) The method according to claim 1, wherein the excitation parameter comprises a fixed codebook excitation component.
3. (Original) The method according to claim 1, wherein the excitation parameter comprises a fixed codebook gain table index.
4. (Original) The method according to claim 1, wherein the excitation parameter comprises a fixed codebook gain parameter.
5. (Original) The method according to claim 4, further comprising the step of multiplying the fixed codebook gain parameter by a scaling factor.
6. (Original) The method according to claim 5, wherein the scaling factor is a constant value.
7. (Original) The method according to claim 6, wherein the constant value is approximately 0.3.
8. (Original) The method according to claim 1, wherein the excitation parameter comprises a fixed codebook gain component and an adaptive codebook gain component.

Serial No. 10/674,450

9. (Original) The method according to claim 8, further comprising the step of multiplying the fixed codebook gain component by a scaling factor.
10. (Original) The method according to claim 9, wherein the scaling factor is a variable scaling factor.
11. (Original) The method according to claim 10, further comprising the step of computing the variable scaling factor as a function of the adaptive codebook gain component.
12. (Previously presented) A method for estimating noise in a speech signal in a communications network, wherein the speech signal is encoded and transported through the network as a bit stream, the method comprising:
 - partially decoding the bit stream to obtain a fixed codebook excitation component and an adaptive codebook excitation component corresponding to the encoded speech signal; and
 - estimating a noise level of the speech signal using the fixed codebook excitation component and the adaptive codebook excitation component obtained directly from the partially decoded bit stream.
13. (Original) The method according to claim 12, further comprising the step of scaling the fixed codebook excitation component according to a constant value.
14. (Original) The method according to claim 12, further comprising the step of scaling the fixed codebook excitation component as a function of the adaptive codebook excitation component.
15. (Previously presented) An apparatus for processing a speech signal, the apparatus comprising:
 - a decoder for partially decoding a bit stream corresponding to an encoded speech signal to extract an excitation parameter; and
 - a noise estimator operable to estimate a noise level in the speech signal using the excitation parameter that is directly obtained from the partially decoded bit stream.

Serial No. 10/674,450

16. (Original) The apparatus according to claim 15, wherein the excitation parameter comprises a parameter selected from the group consisting of a fixed codebook excitation component, a fixed codebook gain table index, and a fixed codebook gain parameter.

17. (Original) The apparatus according to claim 15, further comprising a multiplier element operable to multiply the excitation parameter by a scaling factor.

18. (Original) The apparatus according to claim 17, wherein the scaling factor is a constant value.

19. (Original) The apparatus according to claim 15, wherein the excitation parameter comprises a fixed codebook gain component and an adaptive codebook gain component.

20. (Original) The apparatus according to claim 19, further comprising a multiplier element operable to multiply the fixed codebook gain component by a scaling factor.

21. (Original) The apparatus according to claim 20, wherein the scaling factor is variable as a function of the adaptive codebook gain component.